D24S
VOICE GATEWAY

With a simple and economical way to help legacy telephones, fax machines, and PBXs interconnect with an IP network, Bittel's 24 or 48 ports analog FXS gateway enables call center and multi-branch enterprises to process powerful, versatile, and efficient VoIP solutions with unparalleled cost advantages. Connected between a PBX, LAN, or WAN, the 24 or 48 ports FXS VoIP Gateway converts analog PSTN messages into a format suitable for transmission over standard IP networks.

24 PORTS (FRONT)

Designed for voicemail and unified messaging applications, the 24 or 48 Ports Analog VoIP Gateway D24S has a 10/100/1000M (optional) Base-T Ethernet connection for connecting legacy PBX to a LAN. The analog loop start functionality supports integration via in-band signaling (DTMF or FSK), serial protocols, as well as T.38 for fax transmissions over IP (FoIP).

(BACK)

BENEFITS

- High-performance VoIP connectivity for SMBs.
- Voice optimization to ensure better user experiences.
- Enhanced call routing ability with high voice quality.
- Easy to install, configure, and maintain.
- Support IPv4 and IPv6 international networks.
- Data/voice/management VLAN and more.
- Build-in firewall and access rules.
- Support SNMP/TR069/Auto-Provision.
- Cloud-based management and bandwidth optimization.
- Support SIP, MGCP, or other customizable protocols.
- Primary/Backup SIP Servers.
- Flexible routing and manipulation.

KEY FEATURES

- Completely non-blocking architecture and Scalable System.
- Easy integration with existing telephony interfaces.
- Open-standard SIP support and register to multiple SIP proxy servers.
- Make and receive IP calls from analog extensions.
- Call budgeting is based on the allocated amount, minutes, and call count.
- Manageable-based call routing TDP-IP/IP-TDM.
- Restrict unwanted calls with a list of denied numbers.
- Real-time call record sends to CDR server.
- Caller ID presentation and restriction.
- Hotline extension setting.
- Web-based remote administration.
- Consol access via Telnet, SSH.

PART#: D24S

- Support 24 or 48 FXS Ports, Field Approved Globally
- Superior Voice Quality by Designated DSP Chipsets
- User-Friendliness and Web-based Administration

TECHNICAL SPECIFICATIONS:

PHYSICAL INTERFACE

Phone Interface: RJ21 Amphenol connector available as well Ethernet Interface: 2* RJ-45 10/100Mbps Base-T Ethernet, Female RJ-45 1000M LAN/WAN available for some product models while required.

SESSION CAPACITY

24 or 48 SIP channels (D24S)

24 or 48 FXS channels (D24S)

CONNECTIVITY

Dial Mode: DTMF and Pulse

Pulse: 10 and 20PPS

Caller ID:DTMF/FSK

Max Cable Length:5KM

Reversed Polarity

OpenVPN

VOIP PROTOCOLS

TLS / SRTP

OpenVPN

SIP V2.0 (RFC 3261, 3262, 3264)

IMS/3GPP

SDP

REFER (RFC 3515)

RTP/RTCP

STUN (RFC3489)

ARP/RARP (RFC 826/903)

SNTP (RFC 2030)

DHCP/PPPoE

TFTP/HTTP/HTTPS

DNS/DNSSRV (RFC 1706/RFC2782).

VLAN802. 1P/802.1Q.

CALL & ROUTING

Port Groups

IP Trunks

Primary and Secondary SIP Account

24 or 48 Inbound/OutboundRouting

Number Manipulation

Digit maps

TDMtoIPorIPtoTDM

IP load balancing

IP fault tolerance

VOICE CAPABILITY

G.711A/U law, G.723.1, G.729A/B,G.726,iLBC,AMR

Comfort Noise Generation(CNG)

Echo Cancellation(G.168)

DTMF mode: Signal/RFC2833/INBAND

Silence suppression with comfort noise

G.168 automatic echo cancellation

Call Progress Analysis (CPA), including Positive Voice Detection, Positive

ANSWERING

Machine Detection (PAMD), DTMF detection, and fax tone detection

Manageable based call routing TDP-IP/IP-TDM.

Restrict unwanted calls with list of denied numbers.

Voice Activity Detection (VAD)

Adaptive (Dynamic) Jitter Buffer

Programmable Gain Control

Hook Flash

FOIP PROTOCOL & FAXING

T.38 FoIP: transcode fax from T.30 fax protocol (supporting V.17) modulation schemes

NETWORK CAPABILITY

TStatic IP, PPPoE, DHCP Client IPv4, IPv6

Static/dynamic ARP DIFFServ, ToS

NAT (Rout and Bridge)+ MAC Address Clone Static routing+

Built-in Firewalls

QoS, Traffic Shaping

Voice/Data/Management Vlan

TECHNICAL SPECIFICATIONS CONTD:

MAINTENANCE & UPGRADING

SNMP/TR069.

Auto Provision

Action URL

Digit map

Web/Telnet. ACL

Configuration Backup/Restore

Bandwidth Optimization

Routing Rules based Prefixes

Firmware Upgrade via WEB

Syslog and CDR.

Access Rule list.

Network Capture

Outward Test(GR909).

Automatic Time Synchronization

IVR local Maintenance.

Cloud-based Management

Caller/Called Number Manipulation

Open-standard SIP support and register to multiple SIP proxy servers.

CONFERENCING RESOURCE

Call budgeting based on allocated amount, minutes and call count

Complete non-blocking architecture and Scalable System

Hotline extension setting

Support 3-Way and Multi-Way Conferencing

ENVIRONMENT & POWER

Power Supply: 100-240V, 50-60Hz+

Power Consumption: Approximately 50W

Temperature(Operation):0 °C ~ 45°C

(Storage): -20 ~85°C

Humidity: 10%-90% No condensation.

Operatingtemperaturerange:-10 °C~55°C

PHYSICAL DIMENSION

L*W*H 440(mm)*202(mm)*44(mm)

Weight Approximately 5.95ibs(about 2.7kg)

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APPLICATION CAPABILITIES

Call waiting

Blind Transfer

Attend Transfer

Call forward on Busy

Call forward on No Reply

Unconditional Call Forward

HotlineCall hold

DND

Call Pickup

3-way conference

Voicemail

WARRANTY/CERTIFICATIONS

1 year warranty. CE, FCC

Broadsoft, Elastix, Asterisk, Teams and other UC platform